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Marash et al.

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(54) **SYSTEM, METHOD AND APPARATUS FOR CANCELLING NOISE**

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(58) **Field of Search** 704/270, 500, 704/233, 200, 201, 205, 226, 227, 228, 211, 216; 379/22.08, 392.01, 3, 406.01, 406.12, 406.13, 406.14, 406.05

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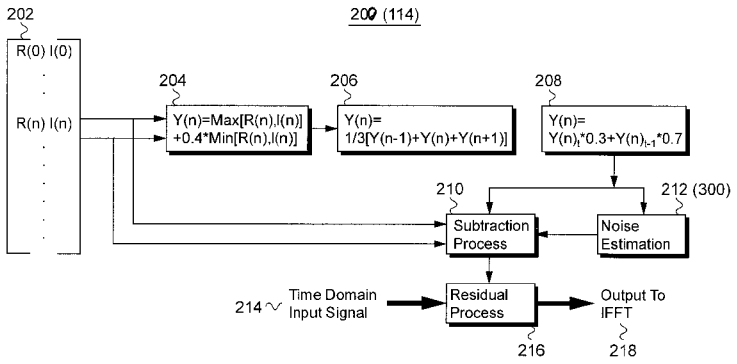
(57) **ABSTRACT**

A threshold detector precisely detects the positions of the noise elements, even within continuous speech segments, by determining whether frequency spectrum elements, or bins, of the input signal are within a threshold set according to current and future minimum values of the frequency spectrum elements. In addition, the threshold is continuously set and initiated within a predetermined period of time. The estimate magnitude of the input audio signal is obtained using a multiplying combination of the real and imaginary part of the input in accordance with the higher and lower values between the real and imaginary part of the signal. In order to further reduce instability of the spectral estimation, a two-dimensional smoothing is applied to the signal estimate using neighboring frequency bins and an exponential average over time. A filter multiplication effects the subtraction thereby avoiding phase calculation difficulties and effecting full-wave rectification which further reduces artifacts. Since the noise elements are determined within continuous speech segments, the noise is canceled from the audio signal nearly continuously thereby providing excellent noise cancellation characteristics. Residual noise reduction reduces the residual noise remaining after noise cancellation. Implementation may be effected in various noise canceling schemes including adaptive beamforming and noise cancellation using computer program applications installed as software or hardware.

47 Claims, 10 Drawing Sheets



Noise Processing